AEQ PHOENIX AUDIOCODECS. APPLICATION NOTE 0-B

Connecting AEQ Phoenix Mobile units through Internet via exclusive DSL or cable modem with DHCP router

(Adapted to Phoenix Mobile user interface)

The most convenient and reliable way to connect an IP codec to a remote one through Internet is by means of a dedicated, exclusive Internet access (DSL, cable modem) as described below. No other equipments (PC, etc) are connected to this same access.

The modem itself usually has an embedded DHCP server that automatically assigns valid IP address / mask, gateway and DNS server to the IP system(s) connected to it.

This application note also describes the recommended procedure when AEQ SIP Proxy is used, where both equipments involved in the communication are registered, easing the task of finding the remote IP address when it is variable or unknown.

This document shows the specific steps to make this connection possible in the most general way possible, although some verification procedures have been omitted in order to ease the reading, and some particular situations may be easier to solve that the presented one.

For a detailed explanation of every step, and for details on how to configure other options that may be more convenient for other cases, the reading of the rest of application notes, as well as User Manual's chapter 4, is recommended.

1. CONNECTING AND CONFIGURING THE EQUIPMENT

The physical connection will be done to a LAN port.

- Connect the network cable to the LAN1 port at the back of the AEQ Phoenix Mobile. Connect the other end to the Ethernet port to the router LAN port.
- Connect at least a microphone and a pair of headphones to the front audio XLR connectors of the Phoenix Mobile
- Connect the power supply cable to the AEQ Phoenix Mobile. If it is equipped with a battery pack and mains power is not available, you can omit this step.
- Press the ON button at the left side of AEQ Phoenix Mobile. Verify that the main display is illuminated.
- Verify, at the back of the unit, that the green LED at the right of the Ethernet port starts blinking indicating that the IP connection is active.
- Configuring the Ethernet port and assignation of Program channel:

  Make sure that we are selecting "Program" in the main screen (the one representing faders and vumeters). At the lower center area you should read

"Program". Also, by pressing MENU we enter the Wizard menu that should have a PROGRAM legend at the top center area. If COORDINATION appears instead in any of both places, just press Channel key and channel will be shifted to PROGRAM.

Main screen

Wizard screen

From main screen press Menu to access the Wizard menu. Make sure that Select Interface shows "IP (Ether)". Otherwise, please select this.

Highlight the Configure Interface, line using the down arrow key, press Select and configure the following:

DHCP ON

The following parameters will be automatically configured:

Local IP address, for example 192.168.1.22

Gateway IP address, for example 192.168.1.1

Network Mask, for example 255.255.255.0

DNS Server IP address, for example 192.168.1.1

Also set: **SIP Proxy: ON**

2. OTHER POINTS TO CHECK

We are now going to quickly overview the rest of configuration menus where it is recommended to check that every parameter is correctly configured.

Coding Algorithm

At the Wizard menu Sel.Codec should be G722, for a start, although you can change this at a later stage for better quality. In order to change the coding algorithm by another pre-selected one, just select the Sel.codec line, press Select and select one within the new window shown. In order to choose the particular coding modes in each list shown in the Sel.codec menu, go to Menu → ADVANCED → 6. Link profiles → Select → Options.

The audio routing will be most likely correct by default to send Mic 1 audio to the transmission and send the received stream to HP1 headphone output. You can, however, make sure of this from Menu → ADVANCED → 3. Audio Settings → Select.

Configure the rest of parameters after reading the manual according to your particular needs.

PROXY SIP Configuration

You can make sure that this is correct from the main screen: Menu → ADVANCED → 5. Communications → 3. SIP config → 1. Parameters:

Check that User Name is filled with a user identifier that is previously registered at the AEQ SIP server (for example: phoenix_123).

STUN Protocol

2 Activate ON

This is important and recommended, as it will ease connecting to other equipment that is behind a router in a local network.

3 Server: At the moment of writing this document, the following STUN server is recommended: "**stun.voxgratia,org**" (IP=83.103.82.85)

If the DNS Server field in the communications menu hasn't been configured, you must substitute the STUN name by its IP.

4 Port : **3478**

Press "back" key in order to go to 3. SIP config → 2. Proxy and check:

1 Activate ON

2 Address: sip.aeq.es.

If the DNS Server field in the communications menu hasn't been configured, you must substitute the SIP server name ("sip.aeq.es") by its IP, 82.165.157.20 at the moment of writing this document.

3 USER must have the same text value as USER NAME from the former menu, followed by "@sip.aeq.es", for example: phoenix_123@sip.aeq.es

4 Check that a correct password is typed in. ("aeq" by default)

5 Check that REALM is "sip.aeq.es".

6 Check that DOMAIN is "sip.aeq.es".

7 Check that EXPIRES = 60

Press back to go to 3. SIP config, then select 4. Call Settings and check that Media PORT is 5004.

3. REGISTERING AT THE AEQ SIP SERVER

If everything is correctly configured, after these steps the unit should have registered at the AEQ SIP server and the top line of every screen should display the current time.

In case it is not registered, if the equipment power is cycled, the sequence of legends ad the top line should be:

<Current time>

Initializing

Finding Nat Type

Register Error

If this happens, just double-check that the configuration is correct and, if necessary, ask one of our technical assistance services for support.

If the unit registers correctly, the sequence will be:

<Current time>

Initializing

Finding Nat Type

<Current time>

4. SENDING A TEST CALL TO PHOENIX MASTER

We are going to go through the correct configuration of the system and prepare it to make a test call to a unit called phoenixMaster@sip.aeq.es that AEQ maintains for our

customers and that continuously transmits audio at the public IP address 212.170.163.189 **.**

Check that a set of headphones are connected to the front HP1 connection.

Check that AEQMaster_IP entry is listed in the Call Book and launch the call. In order to do so…

From main screen: press Menu. Observe the "Select contact" line. If it is:

Select contact: AEQMaster_IP

…everything is ready and the call is sent only by pressing the green key.

If the line "Select contact" presents a different name than **AEQMaster_IP** at its right, just select this line and press "select" key. The call book will be presented. Just select **AEQMaster_IP** from the list, press OK to close the list and then finally press the green key.

If the contact was not already created in the Call Book, you can create it from the menu:

Main screen → Menu → ADVANCED → 2. Contacts. With the help of the user manual, create a new contact with the following parameters:

Contact: **AEQMaster_IP**

Module Type: IP

URI: **phoenixMaster@sip.aeq.es**

(when PROXY SIP = ON , the unit understands that, if "@sip.aeq.es" is omitted, the destination is registered in the same server as this unit.
Speed Dial (n): (leave it as it was).

- The top line at the screen displays (usually very fast) the status of the call, and the different phases it goes through: Calling, Synchronizing and finally **Connected.**
- The second line should show **AEQMaster_IP**
- The third line should show **G722** or the current audio coding scheme.

If the remote unit is busy, the top line will show "**Busy (486)"**

- Verify that the LED at the side of "TX PROG SYNC" key is lit green, as an indication that the communication has been successfully established and synchronized. If the key above HP1 is activated (illuminated) and the

corresponding fader is open, the audio received from Phoenix Master should be heard at the headphones.

Once the connection is OK, you can retry as many times as needed varying the selected audio coding scheme in order to get the best possible quality allowed by the network (Wizard Menu → Sel Codec.)

The list of available coding algorithms can be created in Menu → ADVANCED → 6. Link Profiles → Select

5. SENDING A CALL TO ANOTHER PHOENIX OR N/ACIP COMPATIBLE CODEC

The test call sent to **AEQMaster_IP** has been automatically answered. In order to make a manual call to other codec, you must configure the second one the same way as the first unit, provided that it is also connected directly to a DHCP modem-router. If it is not, just follow the corresponding Internet access (AN0C to AN0F) application note.

Once the reporter equipment connects properly to **AEQMaster_IP**, you can create for convenience new entries in the call book: Menu → ADVANCED → 2. Contacts , with the name and URI of the remote unit, the same way it was done for Phoenix Master in the previous chapter, and then easily make calls to them.

You can of course also send the calls without having previously created a Call Book entry from the main screen, by pressing Menu to go to the Wizard. Select the Call to line, press Select and “URI” is selected. Then type the desired destination URI in the line below (select “Enter URI”, then press OK and type the URI in). For this communications mode based on a SIP server, you only need to introduce the name of the unit (for example, “Phoenix_123”).

The call indications (optical and also acoustical, if enabled) will be activated at the destination codec, that will either wait for the user to answer the call or automatically off-hook (if this mode is enabled), and then establish the connection. The LED next to TX PROG should become green lit and audio should be heard through the headphones.

The three first screen lines should be:

Connected

<equipment name>@sip.aeq.es

<coding scheme>

For more information and other configuration options, please read the User Manual and Application Notes corresponding to other scenarios. If you still have doubts, please don't hesitate to contact one of our assistance services:

sat@aeq.es

support@aeqbroadcast.com